

Adaptive Channel Estimation For SCFDMA System

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ABSTRACT

Third generation partnership project (3GPP) long term evolution (LTE) uses single carrier frequency division multiple access (SC-FDMA) in uplink transmission and orthogonal frequency division multiple access (OFDMA) scheme for the downlink. A variable step size based least mean squares (LMS) algorithm is formulated for a single carrier frequency division multiple access (SC-FDMA) system channel estimation (CE). The weighting coefficients on the channel condition can be updated using this unbiased CE method. Channel and noise statistics information are not essential. Rather, it uses a phase weighting scheme to eliminate the signal fluctuations due to noise and decision errors. The convergence towards the true channel coefficient is guaranteed.

Keywords: Channel estimation, SC-FDMA, LMS

I. INTRODUCTION

Wide demand on high data rates in wireless communication systems has arisen in order to support broadband services. The Third Generation Partnership Project (3GPP) Long Term Evolution (LTE) radio access standard provides peak data rates of 75 Mb/s on the uplink and 300 Mb/s on the downlink. In LTE standard orthogonal frequency-division multiple accesses (OFDMA) is used on the downlink. This supports different carrier bandwidths (1.25–20 MHz) in both frequency-division duplex (FDD) and time-division duplex (TDD) modes. In OFDMA each user is provided with a unique fraction of the system bandwidth. OFDMA combines scalability, multipath robustness, multiple-input multiple-output (MIMO) compatibility [1], thereby making it adaptive for wideband wireless accessibility.

OFDMA, being sensitive to frequency offset and phase noise, accurate frequency and phase synchronization is needed. In addition, OFDMA is characterized by a high transmit PAPR, and for a given peak power limited amplifier this results in a lower mean transmit level. For these reasons, OFDMA is not well suited to the uplink

transmission. Hence, LTE proposed, single carrier FDMA (SC-FDMA), also known as discrete Fourier transform (DFT) pre coded OFDMA, for the uplink. PAPR reduction in SCFDMA is motivated by a desire to increase the mean transmit power, improve the power amplifier efficiency, increase the data rate, and reduce the bit error rate (BER) and energy consumption [4]. SC-FDMA ensures high data rate transmission, utilizing single carrier in frequency domain equalization. In algorithm is proposed for LTE uplink channel impulse response (CIR) knowledge of the channel.

2.1 Baseband System Model

At the transmitter side, a baseband modulator transmits the binary input to a multilevel sequences of complex numbers $m_1(q)$ in one of several possible modulation formats including binary phase shift keying (BPSK), quaternary PSK (QPSK), 8 level PSK (8-PSK), 16-QAM, and 64-QAM. These modulated symbols perform an N-point discrete Fourier transform (DFT) to produce a frequency domain representation. The DFT followed by IDFT in a distribution-FDMA (DFDMA) or localization-FDMA (LFDMA) subcarrier mapping setup operates as efficient implementation to an interpolation filter.

In distributed subcarrier mode, the outputs are allocated equally spaced subcarriers, with zeros occupying the unused subcarrier in between. While in localized subcarrier mode, the outputs are confined to a continuous spectrum of subcarriers. Except the above two modes, interleaved subcarrier mapping mode of FDMA (IFDMA) is another special subcarrier mapping mode [12], [13]. The difference between DFDMA and IFDMA is that the outputs of IFDMA are allocated over the entire bandwidth; whereas the DFDMA's outputs are allocated every several subcarriers.

2.2 Channel Model

Channel model is a mathematical representation of the transfer characteristics of the physical medium. These models are formulated by observing the characteristics of the received signal. According to the documents from 3GPP, in the mobile environment, radio wave propagation can be described by multipaths which arise from reflection and scattering. If there are l distinct paths from transmitter to the receiver, the impulse response of the wide-sense stationary uncorrelated scattering (WSSUS) fading channel can be represented as

$$w(\tau, t) = \sum_{j=0}^{l-1} w_j(t) \delta(\tau - \tau_j)$$

where fading channel coefficients $w_j(t)$ are the wide sense stationary i.e. $w_j(t) = w(m; j)$, uncorrelated complex Gaussian random paths gains at time instant t with their respective delays τ_j , where $w(m; j)$ is the sample spaced channel response of the l^{th} path during the time k , and $\delta(\bullet)$ is the Dirac delta function. Based on the WSSUS assumption, the fading channel coefficients in different delay taps are statistically independent. And has an autocorrelation function given by

$$E[w(k, j)w^T(n, j)] = \sigma_w^2(j) J_0(2\pi f_d T_f (k - n))$$

where $w(n; j)$ is a response of the l^{th} propagation path measured at time n , $\sigma_w^2(J)$ denotes the power of the channel coefficients, f_d is the Doppler frequency in Hertz, T_f is the symbol duration in seconds, and $J_0(\bullet)$ is the zero order Bessel function of the first kind

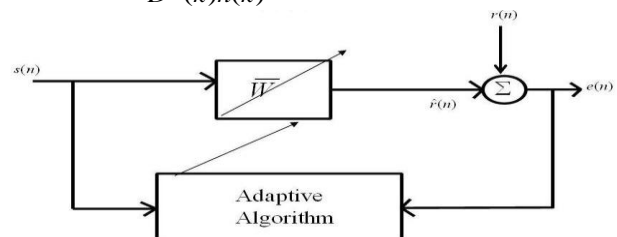
III. PROPOSED ADAPTIVE CE ALGORITHM

The signal $s(k)$ is transmitted via a time-varying channel $w(k)$, and corrupted by an observation noise $n(m)$ before being detected in a receiver. The signal received at time index k is

$$\begin{aligned} r(k) &= s_1(k-1)w_1(m) + \dots + s_l(k-l)w_l(k) + n(k) \\ &= \sum_{j=1}^l s_j(k-j)w_j + n(k) \\ &= S^T(k)W(k) + n(k) \end{aligned}$$

Where $s_j(k-j)$, $j=1,2,\dots,l$ are transmitted signal vectors at time m , l is the distinct paths from transmitter to the receiver, $w(m)$ is the channel coefficients at time m , and $n(k)$ is the noise with zero mean and variance σ^2 . After processing some intermediate steps (synchronization, remove CP, DFT, and de mapping), the decision block reconstructs the detected signal to an approximate modulated signal and its phase. The output $y(k)$ of the adaptive filter is expressed as

$$\begin{aligned} y(k) &= d_1(k-1)h_1(k) + \dots + d_l(k-l)h_l(m) \\ &= \sum_{j=1}^l d_j(k-j)h_j(k) \\ &= D^T(k)h(k) \end{aligned}$$



where $d_j(k-j)$, $j=1,2,\dots,l$ are detected signal vectors at time

$$k \dots D(k) = \text{diag} [d_1(k-1), d_2(k-2), \dots, d_l(k-l)]$$

In this problem formulation, the ideal adaptation procedure would adjust $w_j(k)$ such that $w_j(k) = h_j(k)$ as $k \rightarrow \infty$. In practice, the adaptive filter can only adjust $w(k)$ such that $y(k)$ closely approximates desired signal over time. Therefore, the instantaneous estimated error signal needed to update the weights of the adaptive filter is

$$\begin{aligned} j(k) &= p(k)e^T(k)e(k) \\ e(k) &= r(k) - y(k) \\ &= r(k) - D^T(k)h(k) \end{aligned}$$

This priori error signal, $e(k)$ is used to minimize the estimator error by adaptive updating of filter weights.

3.1 Least Mean Squares (LMS) Algorithm

The LMS algorithm is based on the stochastic gradient and is given by [10]

$$e(k) = S^T(k)w(k) + z(k) - S^T(k)h(k)$$

$$h(k+1) = h(k) + \eta s(k)e(k)$$

where η is step size, $S(k)$ is the transmitted diagonal matrix at sampling time k , $h(k)$ is the adaptive filter coefficient, and $e(k)$ is the estimation error. The filter coefficients are updated using an estimate of the cost function gradient $\nabla_{s(k)e(k)}$.

In all practical applications, the signals involved might be corrupted by noise. When the noise is present in the received sequence, interference will also in the coefficients adaption process through the term $\nabla_{s(k)e(k)}$. As a result, where the distribution of the noise is highly impulsive, the LMS scheme might have low convergence and lower steady state MSE performance. The step size parameter, η determines the convergence rate of the algorithm and higher value provides faster convergence. However, if η exceeds certain bound then the algorithm will diverge. As the bound on η is not known a priori and is dependent on the various statistics

3.2 Normalized LMS (NLMS) Algorithm

The main problem of the LMS CE algorithm is that it is sensitive to the scaling of its input signals. This makes it very hard to choose η that guarantees stability of the algorithm. The NLMS is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input signal. The NLMS algorithm can be summarized as [7]:

$$h(k+1) = h(k) + \eta e(k) \frac{S^T(k)S(k)}{\|S(k)\|^2} S(k)$$

when a constant scalar step size is employed in the LMS or NLMS algorithm, there is a trade off among the steady state error-convergence towards the true channel coefficients, which avoids a fast convergence when the step size is preferred to be small for small output estimation error. In order to guarantee the algorithm to be convergent, the range of step size is specified but the choice of optimal learning step size has not been appropriately addressed. In order to

deal with these troubles, one key idea is to exploit varying step size during adaptation.

3.3 Recursive Least Squares (RLS) Algorithm

To combat the channel dynamics, the RLS based CE algorithm is frequently used for rapid convergence and improved MSE performance [9]. The standard RLS algorithm is

$$e(k) = S^T(k)w(k) + z(k) - S^T(k)h(k)$$

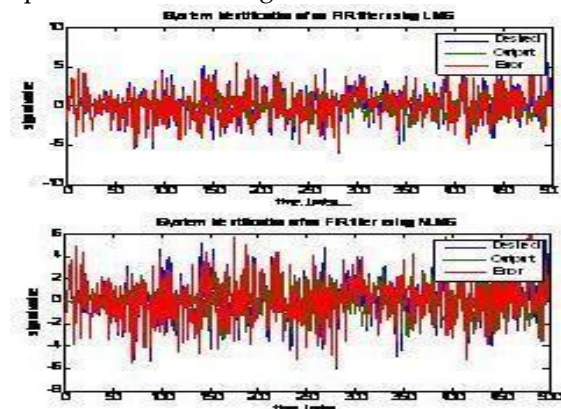
$$R(k) = \lambda^{-1}P(k-1) - \lambda^{-1}R(k)S^T(k)P(k-1)$$

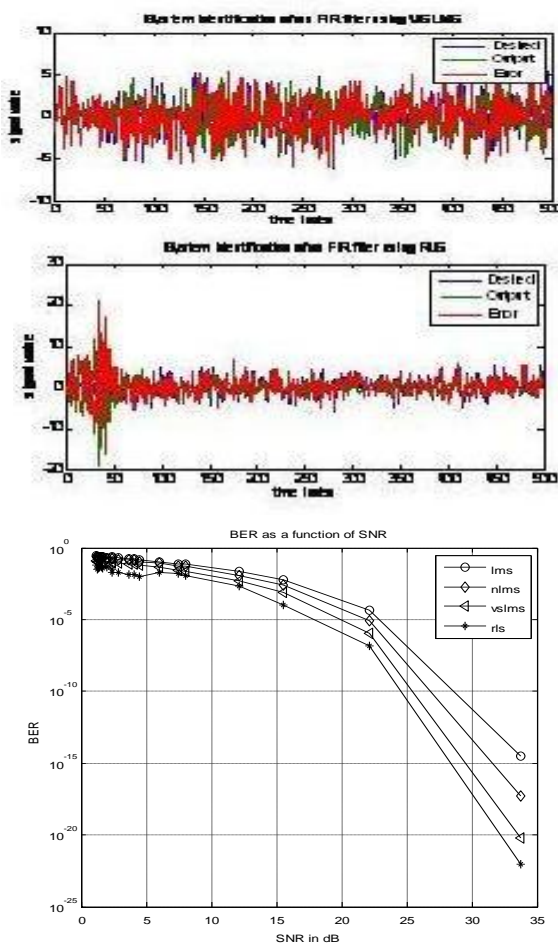
$$h(k+1) = h(k) + S(k)e(k)R(k)$$

Where λ is the exponential forgetting factor with $0 < \lambda < 1$. The smaller value of λ leads to faster convergence rate as well as larger fluctuations in the weight signal after the initial convergence. On the other hand, too small λ value makes this algorithm unstable. Subsequently, it requires best possible forgetting factor such that the estimator error is decreased. Although a lot of modified CE algorithm has been studied on employing adaptive forgetting factor and parallel forgetting factor, the CE performance is severely degraded in highly dynamic fading channel even when the forgetting factor is well optimized [11].

IV. SIMULATION RESULT

The performance of the proposed CE algorithm is compared with the fixed step size LMS algorithm [6], NLMS algorithm [7], VSS-LMS algorithm [8], and RLS algorithm subjected to a Rayleigh fading environment. The simulation parameters are listed in table1. The BER is a significant performance parameter for quality measurement recovered data in wireless communication effect of the proposed CE in terms of BER compared with existing estimators. It is evident that the proposed CE algorithm outperforms the existing.





V. CONCLUSION

A time-varying step size LMS channel estimation scheme is proposed so as to combat channel dynamics and support broadband multimedia access. The weighting coefficients are updated automatically, despite the unavailability of channel information. Besides signals fluctuations due to noise decision errors can be nullified by the phase weighting scheme. Thus the algorithm guarantees convergence towards accurate channel coefficient. Even though, the proposed CE technique requires little bit computational complexity, the advantage of convergence towards true channel coefficient as well as BER performance could be of relevant use in future mobile communications which allow broadband multimedia access, anywhere, and anytime wireless communication.

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